presented above, to continue to adapt frequencies between 0 and 200 Hz during an entire conversation. This is not possible when using a single, time domain convergence parameter. If noise in frequencies below 200 Hz (or in other frequency bins not containing speech) changes during the course of a conversation, the adaptive filter will not be able to adapt with a single convergence parameter because the signal power will indicate that speech is present and will continue to prevent adaptation. However, using the frequency domain approach described herein, convergence on non-speech frequencies can occur DURING speech without adapting to the speech itself.

As mentioned earlier, it is advantageous to combine both of the improvements discussed above to form a third embodiment that provides both robust and optimized control for the dual omni-directional noise canceling microphone. Figure 5 illustrates a block diagram of the combined system incorporating the robust property of creating a passive noise microphone minimal performance (43) with the improved frequency domain adaptive filtering (45) and convergence control (46) discussed above. The reference microphone (41), after being filtered by the single weight adaptive filter (43), is subtracted from (42) the communication microphone (40) to form the minimal performance of the simple active (or passive) noise control microphone. That signal is then used as the communication (or error) signal in the frequency domain adaptive filtering scheme (45) discussed in detail above. As before, the convergence parameters are computed (46) as a function of the spectral power of the output (47) as compared to a stored threshold for each frequency bin.

Having described the invention it is readily apparent that many changes and modification thereto may be made by those of ordinary skill in the art without departing from the scope of the appended claims.

CLAIMS

What is claimed is:

1. An adaptive noise canceling microphone system comprising

a first microphone for generating a first microphone signal containing primarily speech and noise,

a second microphone for generating a second microphone signal containing primarily noise,

a first adaptive filter having a single filter coefficient, generating a first output signal from said first and second microphone signals,

a second adaptive filter having multiple filter coefficients, generating a second output signal from said first output signal and said second microphone signal,

wherein said first output signal is used to update said first adaptive filter and said second output signal is used to update said second adaptive filter, where said second output signal represents primarily speech.

2. A system as in claim 1 wherein the convergence parameter of said first adaptive filter is automatically set to zero after a fixed duration following inception of control so that adaptation of said first adaptive filter ceases to continue.

3. A system as in claim 1 wherein the convergence parameter of said second adaptive filter is automatically switched to zero from a non-zero constant when the said second output signal instantaneously exceeds a fixed, constant, threshold as determined by a comparator.

4. A system as in claim 1 wherein the convergence parameters of said adaptive filters are instantaneously compared to thresholds and updated according to the said first and second output signals.

5. A system as in claim 1 wherein said first microphone is directed toward the speaker's mouth and said second microphone is simultaneously directed away from the speaker's mouth.

6. An adaptive noise canceling microphone control method comprising

a first microphone for generating a first microphone signal containing primarily speech and noise,

a second microphone for generating a second microphone signal containing primarily noise,

a frequency domain adaptive controller generating a control output having a series of stored frequency domain threshold values and a frequency domain comparator, and

an error signal generated by subtracting said first microphone signal from said control output,

whereby the Fourier transform of said control output is compared to said frequency domain threshold values using said frequency domain comparator to generate a series of convergence parameters used to update the frequency domain adaptive controller.

7. A control method as in claim 6 wherein said series of stored frequency domain threshold values is manually entered and stored based on user desired threshold levels.

8. A control method as in claim 6 wherein said series of stored frequency domain threshold values is automatically determined by calculating the Fourier transform of said error signal during a moment in time when no speech is present in said first microphone signal and said Fourier transform of said error signal is stored as the threshold values.

9. A control method as in claim 6 wherein said comparator is implemented using software.

10. A control method as in claim 6 wherein said comparator is implement using hardware.

11. A system as in claim 6 wherein said first microphone is directed toward the speaker's mouth and said second microphone is simultaneously directed away from the speaker's mouth.

12. An adaptive noise canceling microphone control system comprising

a first microphone for generating a first microphone signal containing primarily speech and noise,

a second microphone for generating a second microphone signal containing primarily noise,

a first adaptive filter having a single filter coefficient, generating a first output signal from said first and second microphone signals,

a second frequency domain adaptive filter having multiple filter coefficients, a series of stored frequency domain threshold values, and a frequency domain comparator, generating a second output signal from said first output signal and said second microphone signal,

wherein said first output signal is used to update said first adaptive filter, and said second output signal is used to update said second adaptive filter, where said second output signal represents primarily speech.

13. A system as in claim 12 wherein the convergence parameter of said first adaptive filter is automatically set to zero after a fixed duration following inception of control so that adaptation of said first adaptive filter ceases to continue.

14. A control method as in claim 12 wherein said series of stored frequency domain threshold values is manually entered and stored based on user desired threshold levels.

15. A control method as in claim 12 wherein said series of stored frequency domain threshold values is automatically determined by calculating the Fourier transform of said error signal during a moment in time when no speech is present in said first microphone signal and said Fourier transform of said error signal is stored as the threshold values.

16. A control method as in claim 12 wherein said comparator is implemented using software.

17. A control method as in claim 12 wherein said comparator is implement using hardware.

18. A system as in claim 12 wherein said first microphone is directed toward the speaker's mouth and said second microphone is simultaneously directed away from the speaker's mouth.